(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 809 842 B1

(12) EUROPEAN PATENT SPECIFICATION

(45) Date of publication and mention of the grant of the patent:
02.06.2004 Bulletin 2004/23

(21) Application number: 96905267.9

(22) Date of filing: 29.01.1996

(51) Int Cl.[7]: G10L 21/02, H04R 25/00

(86) International application number:
PCT/US1996/001185

(87) International publication number:
WO 1996/024127 (08.08.1996 Gazette 1996/36)

(54) ADAPTIVE SPEECH FILTER

ADAPTIVER SPRACHSIGNALFILTER

FILTRE VOCAL ADAPTATIF

(84) Designated Contracting States:
AT BE DE DK ES FR GB GR IT NL PT SE

(30) Priority: 30.01.1995 US 380528

(43) Date of publication of application:
03.12.1997 Bulletin 1997/49

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Printed by Jouve, 75001 PARIS (FR)

Description

Field of the Invention

[0001] This invention relates to a method and system for improving the estimates of the spectral components of an information signal, such as speech, from a signal containing both the information signal and noise. The method is particularly suited to implementation on a digital signal processor. The invention also provides the basis for signal enhancement and improved detection of the presence of an information signal.

Background of the Invention

[0002] The spectral components of an information signal are used in a number of signal processing systems including channel vocoders for communication of speech, speech recognition systems and signal enhancement filters. Since the inputs to these systems are often contaminated by noise there has been a great deal of interest in noise reduction techniques.

[0003] The effect of uncorrelated noise is to add a random component to the power in each frequency band.

[0004] Noise free spectral components are required for channel vocoders. In a vocoder the input signal is filtered into a number of different frequency bands and the signal from each band is rectified (squared) and smoothed (low pass filtered). The smoothing process tends to reduce the variance of the noise. Such methods are disclosed in U.S. Patent No. 3,431,355 to Rothauser et al. An alternative approach is disclosed in U.S. Patent No. 3,855,423 to Brendzel et al, in this approach the level of the noise in each band is estimated from successive minima of the energy in that band and the level of the signal is estimated from successive maxima. In U.S. Patent No. 4,000,369 to Paul et al, the noise levels are estimated in a similar fashion and subtracted from the input signals to obtain a better estimate of the speech signal in each band. This method reduces the mean value of the noise.

[0005] Another application of spectral processing is for speech filtering. Weiss et al., in "Processing Speech Signals to Attenuate Interference", presented at the IEEE Symp. Speech Recognition, April 1974, disclose a spectral shaping technique. This technique uses frequency domain processing and describes two approaches - amplitude modulation (which is equivalent to gain control) and amplitude clipping (which is equivalent to a technique called spectral subtraction). Neither the noise estimate nor the speech estimate is updated so this filter is not adaptive. An output time waveform is obtained by recombining the spectral estimates with the original phases.

[0006] An adaptive speech filter is disclosed in U. S. Patent No 4,185,168 to Graupe and Causey. Graupe and Causey describe a method for the adaptive filtering of a noisy speech signal based on the assumption that the noise has relatively stationary statistics compared to the speech signal.

[0007] In Graupe and Causey's method the input signal is divided into a set of signals limited to different frequency bands. The signal to noise ratio for each signal is then estimated in accordance with the time-wise variations of its absolute value. The gain of each signal is then controlled according to an estimate of the signal to noise ratio (the gain typically being close to unity for high signal to noise ratio and less than unity for low signal to noise ratio).

[0008] Graupe and Causey describe a particular method for estimating the noise power from successive minima in the signals, and describe several methods for determining the gain as a function of the estimated noise and signal powers. This is an alternative to the method described earlier in U.S. Patent No. 4,025,721 to Graupe and Causey, which detects the pauses between utterances in the input speech signal and updates estimates of the noise parameters during these pauses. In U.S. Patent No. 4,025,721, Graupe and Causey describe the use of Wiener and Kalman filters to reduce the noise. These filters can be implemented in the time domain or the frequency domain.

[0009] Boll, in "Suppression of Acoustic Noise in Speech using Spectral Subtraction", IEEE Transactions on Acoustics, Speech and Signal Processing. Vol. ASSP-27, No. 2, April, 1979, describes a computationally more efficient way of doing spectral subtraction.

[0010] In the spectral subtraction technique, used by Paul, Weiss and Boll, a constant or slowly-varying estimate of the noise spectrum is subtracted. However, successive measurements of the noise power in each frequency bin vary rapidly and only the mean level of the noise is reduced by spectral subtraction. The residual noise will depend upon the variance of the noise power. This is true also of Weiss's spectral shaping technique where the spectral gains are constant. In Graupe's method the gain applied to each bin is continuously varied so that both the variance and the mean level of the noise can be reduced.

[0011] There are many schemes for determining the spectral gains. One scheme is described by Ephraim and Malah in "Speech enhancement using a minimum mean-square error short-time spectral amplitude estimator", IEEE Transactions on Acoustics, Speech and Signal Processing, Vol. ASSP-32, No. 6, Dec. 1984. This describes a technique for obtaining two estimates of the signal to noise ratio - one from the input signal and one from the output signal. It does not update the estimate of the noise level. The gain is a complicated mathematical function of these two estimates, so this method is not suitable for direct implementation on a digital processor.

[0012] In U.S. Patent No. 5,012,519 to Aldersburg et al the gain estimation technique of Ephraim and Malah is combined with the noise parameter estimation method disclosed in U.S. Patent No. 4,025,721 to Graupe and Causey to provide a fully adaptive system. The mathematical function of Ephraim and Malah is replaced with a two-dimensional lookup table to determine the gains. However, since the estimates of the signal to noise ratio can vary over a very large range, this table requires a large amount of expensive processor memory. Aldersburg et al use a separate voice detection system on the input signal which requires significant additional processing.
[0013] There is therefore a need for an efficient adaptive signal enhancement filter suitable for implementation on an inexpensive digital signal processor.
[0014] There is also a need for a robust noise estimator which can cope with changes in the noise characteristics.
[0015] There is also a need for an efficient signal detection system.

Summary of the Invention

[0016] This invention relates to an improved adaptive spectral estimator for improving the estimates of the spectral components in a signal containing both an information signal, such as speech or music, and noise. According to the invention there are provided a method as set out in claim 1 and a system as set out in claim 10. The improvements relate to a noise power estimator-and a computationally efficient gain calculation method. The adaptive spectral estimator is particularly suited to implementation using digital signal processing. The estimator can be used to provide improved spectral estimates of the information signal and can be combined with a speech or voice recognition system. A further object of the invention is to provide an accurate method for voice detection.

Brief Description of the Drawings

[0017]

Figure I is a diagrammatic view of a system of the prior art.
Figure 2 is a diagrammatic view of a system of the current invention.
Figure 3 is a diagrammatic view of a system for gain modification.
Figure 4 is a diagrammatic view of a system for signal power estimation.
Figure 5 is a diagrammatic view of a system for noise power estimation.
Figure 6 is a diagrammatic view of an information signal detector.

Description of the Preferred Embodiment

[0018] The method is a modified version of that described in U.S. Patent No. 4,185,168 to Graupe and Causey which describes a method for the adaptive filtering of a noisy speech signal. The method is based on the assumption that the noise has relatively stationary statistics compared to the speech signal.
[0019] The input to the filter is usually a digital signal obtained by passing an analog signal, containing noise and the information signal, through high- and low-pass filters and then sampling the resulting signal at a sample rate of at least 8 kHz. The high pass filter is designed to remove low frequency noise which might adversely affect the dynamic range of the filter. The turnover frequency of the high pass filter is less than *f_low,* where *f_low* is the lower limit of the speech band in Hertz. The low pass filter is an anti-aliasing filter which has a turnover frequency of at least *f_high,* where *f_high* is the upper limit of the speech band in Hertz. The order of the low pass filter is determined by the sampling frequency and the need to prevent aliasing.
[0020] The output signal is calculated by filtering the input signal using a frequency domain filter with real coefficients and may be a time series or a set of spectral estimates.
[0021] If the output is a time series then it may be passed to a digital to analog converter (DAC) and an analog anti-imaging filter to produce an analog output signal or it may be used as an input to subsequent signal processing.
[0022] The estimator of the spectral components comprises four basic steps

1. Calculation of the spectrum of the input signal.
2. Estimation of the signal and noise power in each frequency bin within the speech band *(f_low→ f_high* Hz).
3. Calculation of the gains (coefficients) of the frequency domain filter for each frequency bin
4. Calculation of the spectral estimates by multiplying each input spectral component by the corresponding gain.

[0023] This is basically the method of Graupe and Causey which is summarized in Figure 1. Each of the processes is described in detail below.
[0024] The spectral components of the input signal can be obtained by a variety of means, including band pass

filtering and Fourier transformation. In one embodiment a discrete or fast Fourier transform is used to transform sequential blocks of *N* points of the input time series. A window function, such as a Hanning window, can be applied, in which case an overlap of *N/2* points can be used. A Discrete Fourier Transform (DFT) can be used at each frequency bin in the speech band or, alternatively, a Fast Fourier Transform (FFT) can be used over the whole frequency band. The spectrum is stored for each frequency bin within the speech band. For some applications it is desirable to have unequally spaced frequencies - in these applications a Fast Fourier transform cannot be used and each component may have to be calculated independently. In one embodiment the input spectrum, *X*, is calculated as the Fourier transform of the input time series, x, namely

$$X = \text{Fourier transform } \{x, \text{window function}, N\}.$$

[0025] The power in the input spectrum is given by

$$power = \text{modulus squared } \{X\}$$

[0026] Alternatively, a band pass filter may be used, in which case the power may be estimated by rectifying and smoothing the filter output.
[0027] The system of Graupe and Causey is shown in Figure 1.
[0028] The input signal, *x*, is passed to bank of band pass filters. One of these filters 1 is shown in Figure 1. This produces an input component *X*. The power of this component is measured at 2.
[0029] The method requires that estimates are made of the signal power, *signal,* and noise power, *noise.* The noise power is estimated in 3 with a time constant related to the time over which the noise can be considered stationary. The signal is estimated at 4. From these estimates the Wiener filter gain, *W,* is calculated as the ratio of the power in the information signal to the total power. This is done at 5 in Figure 1. For each frequency bin this is

$$W = signal\,/\,(noise + signal).$$

[0030] In the method of Graupe and Causey the Wiener gain, *W,* is directly applied to the corresponding component of the input spectrum. In the unmodified scheme the spectral components of the output are given by multiplying the input component by the gain at 6 in Figure 1. The result is

$$Y = W^*X$$

[0031] If the output time series, *y*, is required it can be calculated by an inverse FFT (or DFT) and the 'overlap-add' method or by summing the components from individual channels using channel summer 7 in Figure 1.
[0032] After each iteration *k* the output block of *N* time points is updated as

$$y_k(1{:}N) = \text{inverse Fourier transform } \{Y,N\}$$

$$y_k(1{:}N/2) = y_k(1{:}N/2) + y_{k-1}(N/2+1{:}N)$$

[0033] The first *N/2* points of y_k are then sent to the DAC or may be used for further processing.
[0034] An improved system of the current invention is shown in Figure 2. The additional features are described below.

Gain Modification

[0035] When the signal to noise ratio is low the direct use of the Wiener gain results in a residual noise which has a musical or artificial character.
[0036] One improvement of the current invention is the use of gain modifier, 8 in Figure 2, which reduces the musical nature of the residual noise. The gain modifier, which is shown in Figure 3, will now be described.
[0037] The instantaneous power of the information signal can be estimated as the product of the instantaneous power and the Wiener gain. This gives an estimate of the instantaneous signal to noise ratio, *snr,* in each frequency

bin obtained by dividing the power by the noise at 10 in Figure 3, and using this to modulate or multiply the Wiener gain at 11. Hence

$$snr = W * (power / noise).$$

[0038] A function of the signal to noise ratio is then calculated at 12. The modified filter gains (coefficients), which are denoted by the vector *C*, are calculated by dividing this function of the signal to noise ratio by the ratio of the power to the noise at 13. This is done for each frequency, so that

$$C = F\{snr\} * (noise / power) = F\{snr\} / (power / noise)$$

where F is a function of a single variable and is therefore well suited to implementation on a DSP as a look-up table or an analytic function. One form of the function *F* is given by

$$F(x) = \begin{cases} x^{1/2}, & x < snr0 \\ x + c, & x \geq snr0 \end{cases}.$$

where *c* and *snr0* are constants. Other forms can used, but it is desirable that the function is approximately linear at high signal to noise ratios. In particular the gain of Ephraim and Malah may be manipulated so that it can be implemented in this form.

[0039] The spectral output, *Y*, that is the estimate of the spectrum of the information signal, is calculated by multiplying the input spectral components by the corresponding modified gains 6 in Figure 2., so that for each frequency

$$Y = C * X$$

Signal Estimation

[0040] Ephraim and Malah describe a method for updating a signal to noise ratio. This method can be modified to give an estimate of the signal power, *signal*. This signal estimator (4 in Figure 2) uses the power in the output signal calculated at 9 in Figure 2. The method is shown in detail in Figure 4 and is given by

$$sig1 = \text{maximum}\{power - noise, 0\}$$

$$sig2 = \text{modulus squared}\{Y\}$$

$$signal = (1\text{-}beta) * sig1 + beta * sig2$$

[0041] The difference between the current total power and the estimate of the noise is calculated at 14. This signal is then half wave rectified at 15. The signal estimate is obtained as a weighted sum 16 of this rectified signal and the power in the output signal The weighting parameter *beta* used in the weighted sum is typically chosen to be greater than 0.9 and less than 1.

Noise Estimation

[0042] The estimates of the noise can be updated during the pauses in the information signal. The pauses can be detected by looking at a weighted sum of the signal to noise components across frequency bins (a uniform weighting may be used). If this weighted sum is below a predetermined threshold, *Smin* say, the noise estimate at each frequency is updated as

$$noise = noise + alpha * \text{maximum}\ \{power - noise,\ 0\}$$

where *alpha* is a parameter which determines the time constant of the estimate. *alpha* is typically chosen to be greater than 0.9 and less than 1.

[0043] An alternative noise estimator may be obtained by using the assumption that the information signal and the noise signal are uncorrelated. The signal power can be estimated from the output components, *Y*, and subtracted from the total power *old_power* from the previous update. That is

$$temp = alpha*(old\_power[f] - signal)$$

$$noise = (1\text{-}alpha) * noise +$$

$$alpha * \text{sign}\ \{temp\}*\text{minimum}\ \{\text{abs}(temp), noise/2\}$$

[0044] This noise estimator is depicted in Figure 5. The difference between the total power and the signal power is calculated at 17, it is then multiplied by *alpha* at 18. The previous noise estimate is multiplied by (1-*alpha*) at 19 and added in 20 to the output of multiplier 18. The noise estimator described above differs from those previously used in that it makes use of the signal estimate. Other forms of noise estimators can be used, including combinations of the above two methods.

Information Signal Detector

[0045] The presence of an information signal can be detected by looking at a weighted sum of the signal to noise components across frequency bins (a uniform weighting may be used). If this weighted sum is above a predetermined threshold, the signal is assumed to contain information. This is shown in Figure 6 the signal to noise ratios are weighted at 22 and then summed at 23 before being passed to the threshold detector 24.

A Particular Embodiment

[0046] One embodiment of the method is described below

```
at each update number k
        X = Fourier transform { x, window function, N }.
        FOR each frequency number f in speech band
                power = modulus squared{ X[f] }
                sig1 = maximum{power - noise[f], 0}
                sig2 = modulus squared{ Y[f]}
                signal = (1-beta) * sig1 + beta * sig2
                W = signal /( noise[f] + signal )
                snr = W * ( power /noise[f] )
                C = F{snr} / ( power /noise[f] )
                temp = alpha*(old_power[f] - signal)
                noise = (1-alpha) * noise +
                        alpha * sign{temp}*minimum{abs(temp),noise/2}
                old_power[f] = power
                Y[f] = C * X[f]
        ENDFOR
    y_k(1:N) = inverse Fourier transform {Y,N}
    y_k(1:N/2) = y_k(1:N/2) + y_k-1(N/2+1:N)
```

[0047] At the end of each iteration, k, the signal $y_k(1{:}N/2)$ provides an estimate of the information signal.

Claims

1. A method for estimating the frequency components of an information signal from an input signal containing both the information signal and noise, said method comprising
filtering the input signal through a set of band pass filters to produce a set of input frequency components, one for each frequency band, and for each frequency component,
calculating the total power in each input frequency component,
estimating the power of the information signal included therein,
calculating a gain for each frequency band as a function of the total power, the estimate of the power in the information signal, and a previous estimate of the noise power,
multiplying the input frequency component by said gain to thereby produce an estimate of the frequency component of said information signal,
estimating a new noise power estimate from the previous noise power estimate and the difference between the total power in the input frequency component and the estimate of the power of the frequency component of said information signal.

2. A method according to claim 1 in which the gain in each frequency band is determined by
estimating a Wiener gain from said previous noise power estimate and the estimate of the power of the information signal,
multiplying said Wiener gain by the ratio of the power of the input frequency component to the estimated noise power to produce an estimate of the signal to noise ratio,

calculating a function of the estimated signal to noise ratio, and

dividing said function of the estimated signal to noise ratio by the ratio of the power of the input frequency component to the estimated noise power to thereby produce a modified gain.

3. A method according to claim 1 or 2, further comprising the step of estimating the overall signal to noise ratio from a weighted sum of the estimated signal to noise ratios in each frequency band.

4. A method according to claim 3, further comprising the step of using said estimated overall signal to noise ratio to determine the presence of an information signal in the input signal.

5. A method according to any of the preceding claims, further comprising the step of recombining the estimates of the frequency components of said information signal to produce a noise reduced output signal.

6. A method according to any of the preceding claims in which the power of the information signal is estimated from a combination of the previous estimate of the frequency components of said information signal and the positive difference between the power in the input frequency component and the noise power estimate.

7. A method according to any of the preceding claims in which the filtering is performed via a Fourier transform.

8. Use of the method of any of the preceding claims for preprocessing signals in a speech or voice recognition system.

9. Use of the method of any of the preceding claims for reducing noise in a signal in a communications system.

10. A system for estimating the frequency components of an information signal from an input signal containing both the information signal and noise, said system comprising

filter means for filtering the input signal through a set of band pass filters (1) to produce a set of input frequency components, one for each frequency band, and for each frequency component,

first calculating means (2) for calculating the total power in each input frequency component,

estimating means (4) for estimating the power of the information signal included therein,

second calculating means (5,8) for calculating a gain for each frequency band as a function of the total power, the estimate of the power in the information signal, and a previous estimate of the noise power,

gain multiplying means (6) for multiplying the input frequency component by said gain to thereby produce an estimate of the frequency component of said information signal whereby said estimating means estimates a new noise power estimate from the previous noise power estimate and the difference between the total power in the input frequency component and the estimate of the power of the frequency component of said information signal.

11. A system according to claim 10 in which the second calculating means comprises

means for estimating a Wiener gain from said previous noise power estimate and the estimate of the power of the information signal,

Weiner multiplying means for multiplying said Wiener gain by the ratio of the power of the input frequency component to the estimated noise power to produce an estimate of the signal to noise ratio,

function calculating means for calculating a function of the estimated signal to noise ratio, and

division means for dividing said function of the estimated signal to noise ratio by the ratio of the power of the input frequency component to the estimated noise power to thereby produce a modified gain.

Patentansprüche

1. Verfahren zum Schätzen der Frequenzkomponenten eines Informationssignals aus einem Eingangssignal, das sowohl das Informationssignal als auch Störgeräusche enthält, wobei das Verfahren folgendes aufweist:

Filtern des Eingangssignals durch einen Satz von Bandpassfiltern, um einen Satz von Eingangs-Frequenzkomponenten, eine für jedes Frequenzband und für jede Frequenzkomponente, zu erhalten,

Berechnen der Gesamtleistung in jeder Eingangs-Frequenzkomponente,

Schätzen der Leistung des darin enthaltenen Informationssignals,

Berechnen einer Verstärkung für jedes Frequenzband in Abhängigkeit von der Gesamtleistung, des Schätzwerts der Leistung im Informationssignal und eines früheren Schätzwerts der Rauschleistung,

Multiplizieren der Eingangs-Frequenzkomponente mit dieser Verstärkung, um dadurch einen Schätzwert der Frequenzkomponente des Infonnationssignals zu erhalten,
Schätzen eines neuen Rauschleistungsschätzwerts aus dem früheren Rauschleistungsschätzwert und der Differenz zwischen der Gesamtleistung in der Eingangs-Frequenzkomponente und dem Schätzwert der Leistung der Frequenzkomponente des Informationssignals.

2. Verfahren nach Anspruch 1, bei dem die Verstärkung in jedem Frequenzband festgestellt wird durch
Schätzen einer Wiener-Verstärkung aus dem früheren Rauschleistungsschätzwert und dem Schätzwert der Leistung des Informationssignals,
Multiplizieren der Wiener-Verstärkung mit dem Verhältnis der Leistung der Eingangs-Frequenzkomponente zur geschätzten Rauschleistung, um einen Schätzwert des Signal-Rausch-Verhältnisses zu erhalten,
Berechnen einer Funktion des geschätzten Signal-Rausch-Verhältnisses und Dividieren dieser Funktion des geschätzten Signal-Rausch-Verhältnisses durch das Verhältnis der Leistung der Eingangs-Frequenzkomponente zur geschätzten Rauschleistung, um dadurch eine modifizierte Verstärkung zu erhalten.

3. Verfahren nach Anspruch 1 oder 2, das weiterhin den Schritt des Schätzens des Gesamt-Signal-Rausch-Verhältnisses aus einer bewerteten Summe der geschätzten Signal-Rausch-Verhältnisse in jedem Frequenzband enthält.

4. Verfahren nach Anspruch 3, das weiterhin den Schritt der Verwendung des geschätzten Gesamt-Signal-Rausch-Verhältnisses zur Feststellung des Vorhandenseins eines Informationssignals im Eingangssignal enthält.

5. Verfahren nach einem der vorhergehenden Ansprüche, das weiterhin den Schritt des Wiedervereinigens der Schätzwerte der Frequenzkomponenten des Informationssignals enthält, um ein rauschreduziertes Ausgangssignal zu erhalten.

6. Verfahren nach einem der vorhergehenden Ansprüche, bei dem die Leistung des Informationssignals aus einer Kombination der früheren Schätzwerte der Frequenzkomponenten des Informationssignals und der positiven Differenz zwischen der Leistung in der Eingangs-Frequenzkomponente und dem Rauschleistungsschätzwert geschätzt wird.

7. Verfahren nach einem der vorhergehenden Ansprüche, bei dem das Filtern über einen Fourier-Transformator stattfindet.

8. Anwendung des Verfahrens eines der vorhergehenden Ansprüche zum Vorverarbeiten von Signalen in einem Spracherkennungssystem.

9. Anwendung des Verfahrens eines der vorhergehenden Ansprüche zur Reduzierung der Störgeräusche in einem Signal in einem Kommunikationssystem.

10. System zum Schätzen der Frequenzkomponenten eines Informationssignals aus einem Eingangssignal, das sowohl das Informationssignal als auch Störgeräusche enthält, wobei das System folgendes aufweist:

einen Filter zum Filtern des Eingangssignals durch einen Satz von Bandpassfiltern (1), um einen Satz von Eingangs-Frequenzkomponenten zu erhalten, eine für jedes Frequenzband und für jede Frequenzkomponente,
eine erste Berechnungseinrichtung (2) zum Berechnen der Gesamtleistung in jeder Eingangs-Frequenzkomponente,
eine Schätzeinrichtung (4) zum Schätzen der Leistung des darin enthaltenen Informationssignals,
eine zweite Berechnungseinrichtung (5, 8) zum Berechnen einer Verstärkung für jedes Frequenzband in Abhängigkeit der Gesamtleistung, des Schätzwerts der Leistung im Informationssignal und eines früheren Schätzwerts der Rauschleistung,
eine Verstärkungs-Vervielfachungseinrichtung (6) zum Multiplizieren der Eingangsfrequenzkomponente mit der Verstärkung, um dadurch einen Schätzwert der Frequenzkomponente des Informationssignals zu erhalten,
wobei die Schätzeinrichtung einen neuen Rauschleistungsschätzwert aus dem früheren Rauschleistungsschätzwert und der Differenz zwischen der Gesamtleistung in der Eingangs-Frequenzkomponente und dem Schätzwert der Leistung der Frequenzkomponente des Informationssignals schätzt.

11. System nach Anspruch 10, bei dem die zweite Berechnungseinrichtung folgendes aufweist:

eine Einrichtung zum Schätzen einer Wiener-Verstärkung aus dem früheren Rauschleistungsschätzwert und dem Schätzwert der Leistung des Informationssignals,
eine Wiener-Vervielfachungseinrichtung zum Multiplizieren der Wiener-Verstärkung mit dem Verhältnis der Leistung der Eingangs-Frequenzkomponente zur geschätzten Rauschleistung, um einen Schätzwert des Signal-Rausch-Verhältnisses zu erhalten,
eine Funktionsberechnungseinrichtung zum Berechnen einer Funktion des geschätzten Signal-Rausch-Verhältnisses, und
eine Dividiereinrichtung zum Dividieren der Funktion des geschätzten Signal-Rausch-Verhältnisses durch das Verhältnis der Leistung der Eingangs-Frequenzkomponente zur geschätzten Rauschleistung, um dadurch eine modifizierte Verstärkung zu erhalten.

Revendications

1. Un procédé pour estimer les composantes fréquentielles d'un signal d'information à partir d'un signal d'entrée contenant à la fois le signal d'information et du bruit, ledit procédé comprenant
   le filtrage du signal d'entrée à travers un ensemble de filtres passe-bande afin de produire une série de composantes fréquentielles d'entrée, une pour chaque bande de fréquences, et pour chaque composante fréquentielle,
   le calcul de la puissance totale dans chaque composante fréquentielle d'entrée,
   l'estimation de la puissance du signal d'information qu'elle comporte,
   le calcul d'un gain pour chaque bande de fréquences en fonction de la puissance totale, de l'estimation de la puissance du signal d'information et d'une estimation précédente de la puissance de bruit,
   la multiplication de la composante fréquentielle d'entrée par ledit gain, afin d'obtenir ainsi une estimation de la composante fréquentielle dudit signal d'information,
   la détermination d'une nouvelle estimation de la puissance de bruit à partir de l'estimation précédente de la puissance de bruit et de la différence entre la puissance totale de la composante fréquentielle d'entrée et l'estimation de la puissance de la composante fréquentielle dudit signal d'information.

2. Un procédé selon la revendication 1 dans lequel le gain dans chaque bande de fréquences est déterminé par
   l'estimation d'un gain de Wiener à partir de ladite estimation précédente de la puissance de bruit et de l'estimation de la puissance du signal d'information,
   la multiplication dudit gain de Wiener par le rapport entre la puissance de la composante fréquentielle d'entrée et la puissance de bruit estimée, afin d'établir une estimation du rapport signal/bruit,
   le calcul d'une fonction du rapport signal/bruit estimé, et
   la division de ladite fonction du rapport signal/bruit estimé par le rapport entre la puissance de la composante fréquentielle d'entrée et la puissance de bruit estimée, afin d'établir ainsi un gain modifié.

3. Un procédé selon la revendication 1 ou 2, comprenant en outre l'étape d'estimation du rapport signal/bruit global à partir d'une somme pondérée des rapports signal/bruit estimés dans chaque bande de fréquences.

4. Un procédé selon la revendication 3, comprenant en outre l'étape consistant à utiliser ledit rapport signal/bruit global estimé pour déterminer la présence d'un signal d'information dans le signal d'entrée.

5. Un procédé selon l'une quelconque des revendications précédentes, comprenant en outre l'étape consistant à recombiner les estimations des composantes fréquentielles dudit signal d'information pour produire un signal de sortie à bruit réduit.

6. Un procédé conforme à l'une quelconque des revendications précédentes dans lequel la puissance du signal d'information est estimée à partir d'une combinaison de l'estimation précédente des composantes fréquentielles dudit signal d'information et de la différence positive entre la puissance de la composante Séquentielle d'entrée et l'estimation de la puissance de bruit.

7. Un procédé conforme à l'une quelconque des revendications précédentes dans lequel le filtrage est effectué à l'aide d'une transformée de Fourier.

8. Utilisation du procédé de l'une quelconque des revendications précédentes pour prétraiter des signaux dans un système de reconnaissance de parole ou de voix.

9. Utilisation du procédé de l'une quelconque des revendications précédentes pour réduire le bruit d'un signal dans un système de communication.

10. Un système pour estimer les composantes fréquentielles d'un signal d'information à partir d'un signal d'entrée contenant à la fois le signal d'information et du bruit, ledit système comprenant

des moyens de filtrage pour filtrer le signal d'entrée à travers un ensemble de filtres passe-bande (1) afin de produire une série de composantes fréquentielles d'entrée, une pour chaque bande de fréquence, et pour chaque composante fréquentielle,

des premiers moyens de calcul (2) pour calculer la puissance totale dans chaque composante fréquentielle d'entrée,

des moyens d'estimation (4) pour estimer la puissance du signal d'information qu'elle comporte,

des deuxièmes moyens de calcul (5,8) pour calculer un gain pour chaque bande de fréquences en fonction de la puissance totale, de l'estimation de la puissance du signal d'information et d'une estimation précédente de la puissance de bruit,

des moyens de multiplication de gain (6) pour multiplier la composante fréquentielle d'entrée par ledit gain, afin de produire ainsi une estimation de la composante fréquentielle dudit signal d'information, lesdits moyens d'estimation procédant à une nouvelle estimation de la puissance de bruit à partir de l'estimation précédente de la puissance de bruit et de la différence entre la puissance totale de la composante fréquentielle d'entrée et l'estimation de la puissance de la composante fréquentielle dudit signal d'information.

11. Un système conforme à la revendication 10 dans lequel les deuxièmes moyens de calcul comprennent ;

des moyens pour estimer un gain de Wiener à partir de ladite estimation précédente de la puissance de bruit et de l'estimation de la puissance du signal d'information,

des moyens de multiplication de Wiener pour multiplier ledit gain de Wiener par le rapport entre la puissance de la composante fréquentielle d'entrée et la puissance de bruit estimée, afin de produire une estimation du rapport signal/bruit,

des moyens de calcul de fonction pour calculer une fonction du rapport signal/bruit estimé, et

des moyens de division pour diviser ladite fonction du rapport signal/bruit estimé par le rapport entre la puissance de la composante fréquentielle d'entrée et la puissance de bruit estimée, afin de produire ainsi un gain modifié.

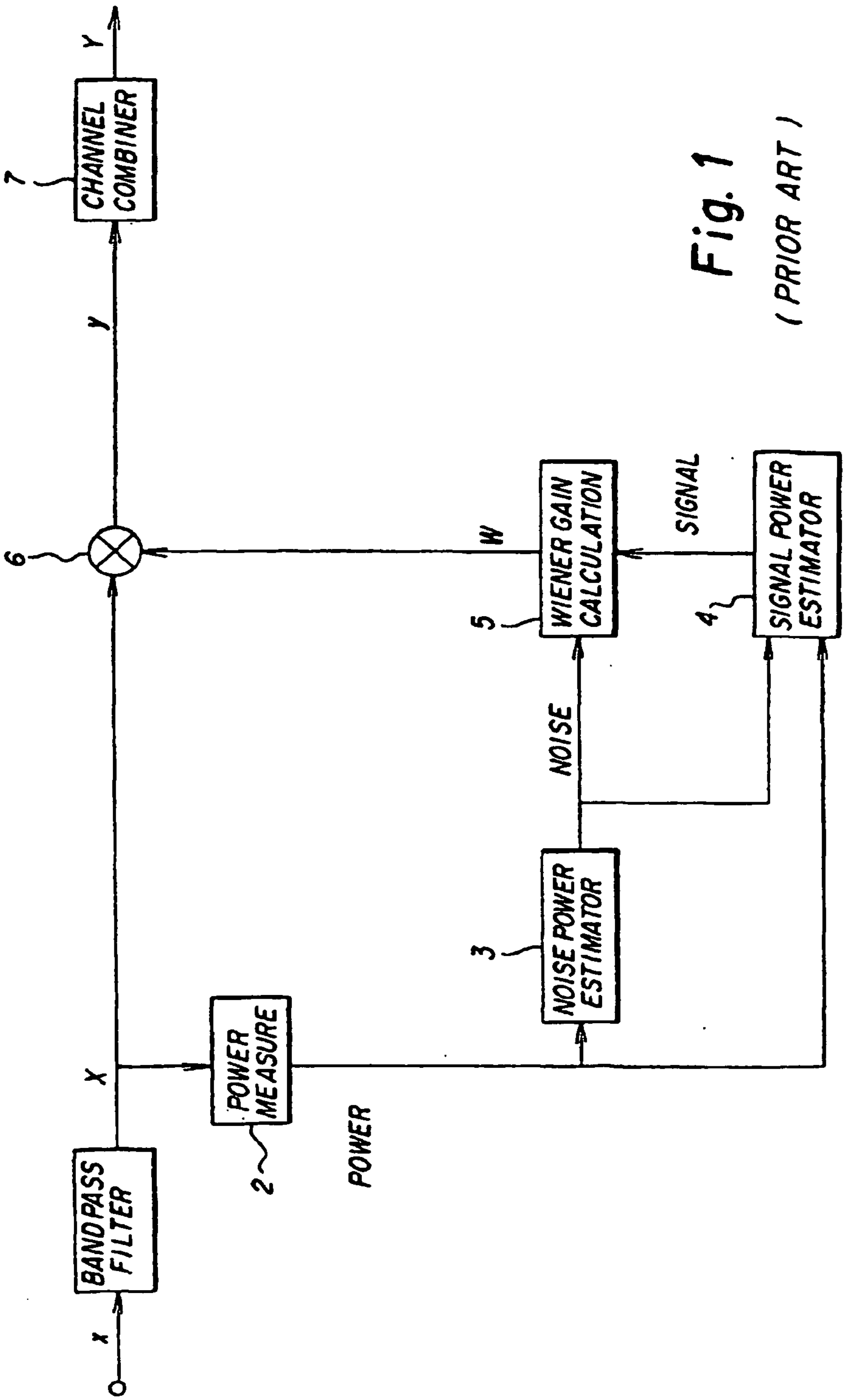

Fig. 1
(PRIOR ART)

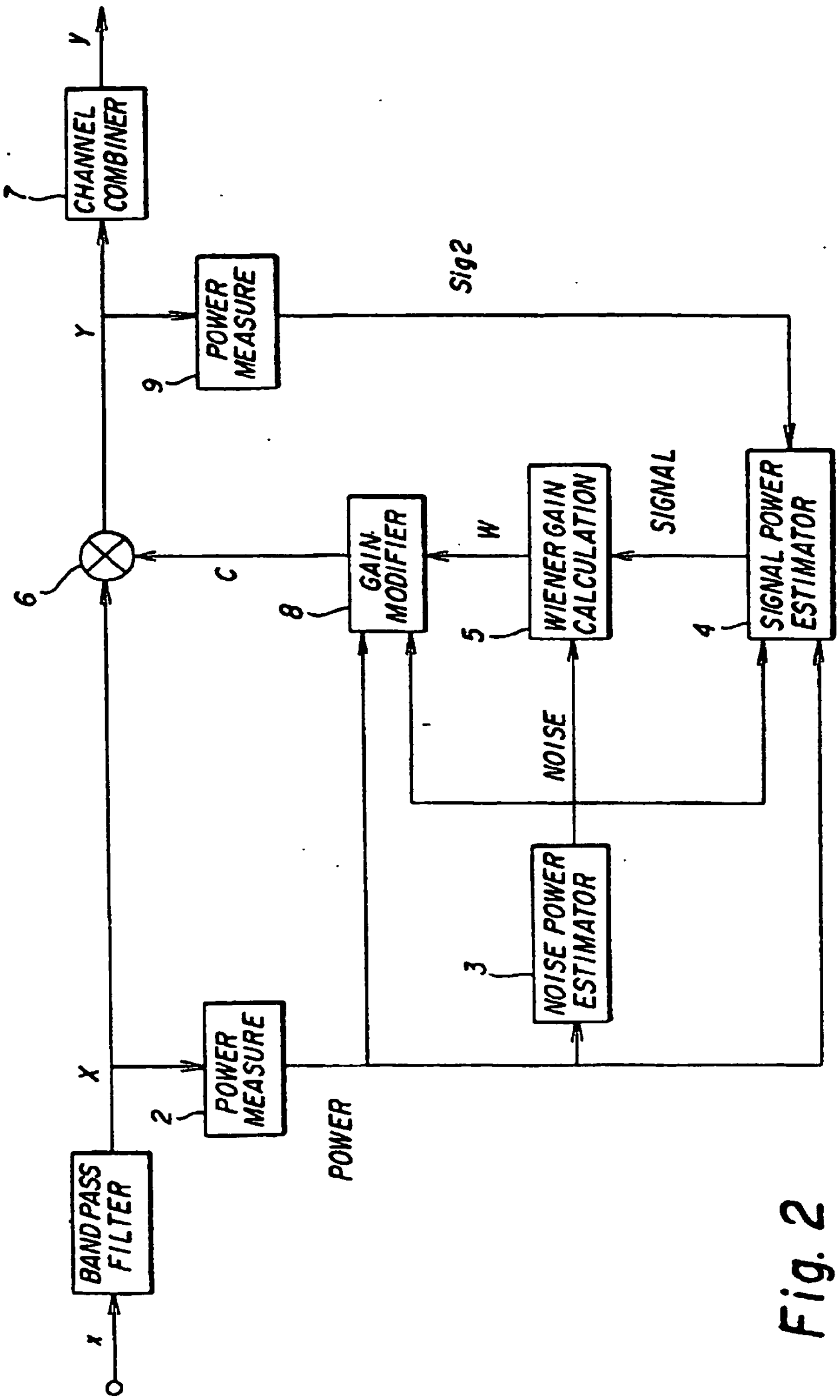
x
BAND PASS FILTER
X
2
POWER MEASURE
POWER
3
NOISE POWER ESTIMATOR
NOISE
6
C
8
GAIN MODIFIER
W
5
WIENER GAIN CALCULATION
SIGNAL
4
SIGNAL POWER ESTIMATOR
Y
9
POWER MEASURE
Sig2
7
CHANNEL COMBINER
y

Fig. 2

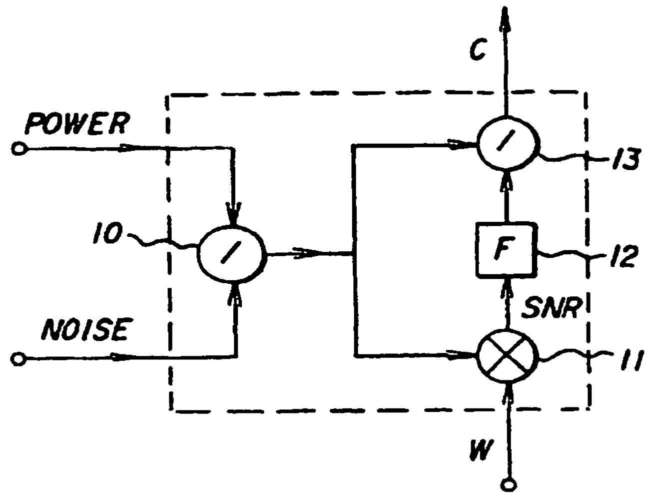
C
POWER
13
10
F
12
SNR
NOISE
11
W

Fig. 3

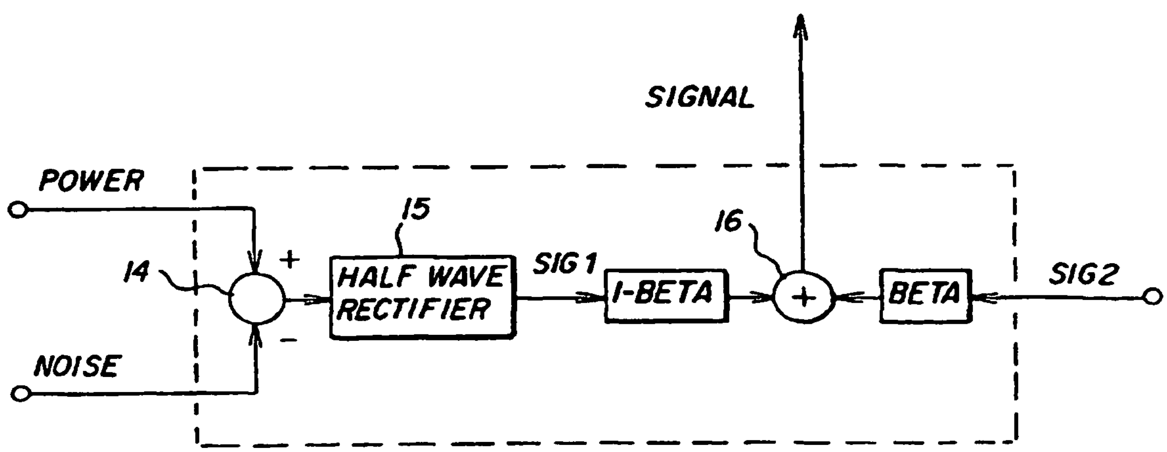
SIGNAL
POWER
15
16
14
+
HALF WAVE
RECTIFIER
SIG 1
1-BETA
+
BETA
SIG2
-
NOISE

Fig. 4

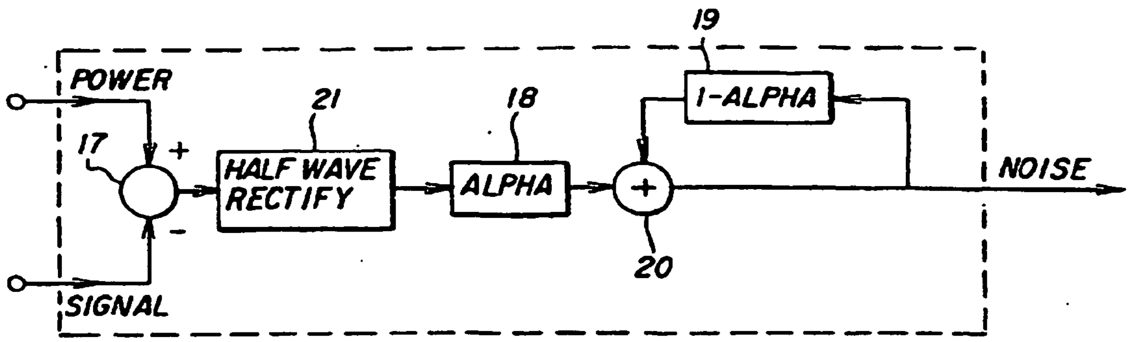
19
POWER
21
18
1-ALPHA
17
+
HALF WAVE
RECTIFY
ALPHA
+
NOISE
-
20
SIGNAL

Fig. 5

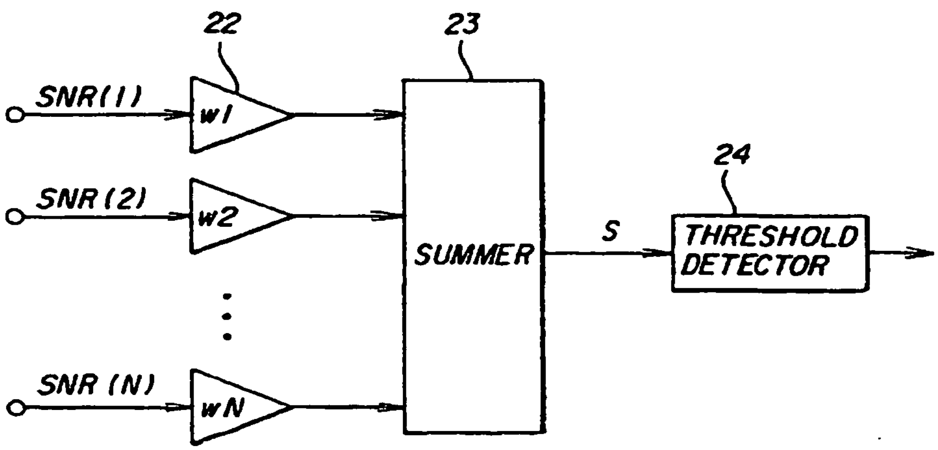
22
23
SNR(1)
w1
24
SNR(2)
w2
S
THRESHOLD
DETECTOR
SUMMER
SNR (N)
wN

Fig. 6